DIGITAL TRANSMISSION SYSTEMS

CONTENTS

1. GENERAL AND HISTORY

1.1 This section provides REA borrowers, consulting engineers and other interested parties with information on digital transmission systems. A brief introduction and history of digital transmission systems is followed by an overview and summary discussion of PCM fundamentals, digital building blocks, and similar topics. A more detailed discussion of digital equipment and application is covered in other sections in the 950 and 960 series of the TE&CM. TE&CM Section 951 contains a glossary of terminology used in digital transmission systems, and is provided as a companion reference for this section and other sections in the 950 and 960 series.

1.2 Digital Equipment: The characteristics of digital transmission and switching equipment are outlined in REA specifications, telephone industry specifications, and in various TE&CM sections. REA Specification PE-60 and PE-64 cover basic digital carrier equipment such as trunk and subscriber terminal equipment, span lines, and automatic protection switching. The detailed specifications for specific installations of digital carrier equipment are covered under REA Forms 397b and 397c. REA Form 522 contains the basic equipment specifications and the detailed installation specifications for digital central office equipment, including remote switching terminals. Digital hardware and software are in the process of "evolving". It will sometimes be necessary to supplement or modify digital system and equipment specifications to keep pace with the latest technology. This is especially the case for some of the high density optical fiber systems, or the more unique distributed subscriber systems for voice and data capability.

1.3 Early digital communications took the form of smoke signals, signal fires, sunlight reflected by mirrors and similar techniques. The telegraph marked the beginning of modern communications where coded (digital) electrical signals were transmitted over wires. The use of light in digital communications has been "rediscovered" with the development of lightwave transmission systems.

1.3.1 Pulse code modulation (PCM) carrier principles were developed by Alec Reeves of ITT in Europe in 1937. At that time, there was no hardware available to convert these concepts into practical systems. In 1956, Bell Labs began the development of the first commercial PCM system. PCM carrier began commercial service in 1962--25 years after Reeves introduced the principles of PCM. The invention of the transistor by Bell Labs in 1947 triggered a large scale electronics revolution. The integrated circuit developed two decades later enhanced the role of digital systems for computers and communication systems.

1.3.2 The Bell System's PCM system was designated as the Western Electric T1 carrier system. Bell developed digital system building blocks or modules, and identified the digital transmission system as T1 span lines and the encoding and decoding PCM terminals as D1 channel banks. From this beginning, a digital hierarchy concept was developed for the trunk network. Independent telephony manufacturers extended the use of PCM carrier into subscriber loop plant before 1970. Digital transmission techniques have now been extended into switching systems. Early Bell efforts were concentrated on digital toll switches, while the Independent manufacturers have concentrated on integrated Class 5 digital switches, remote switches, digital subscriber carrier and digital subscriber line concentrators.

1.3.3 The accelerating development of integrated circuits has further enhanced the role of digital transmission and switching systems in telephony. Major parts of telephone systems can be put on one or several integrated circuits. The information that can be stored on a single integrated circuit continues to expand, providing new markets for these devices. Integrated circuits can now store a digitized voice for recorded messages with no mechanical moving parts to fail. An increase in the use of data transmission by telephone business customers is expected in the future. Digital transmission systems are especially suited for efficient data transmission. Standardization, volume production, power efficiency and smaller size of electronic systems should result in lower costs to telephone companies and their subscribers.

1.4 The term "digital" refers to information that is represented numerically. Digital transmission systems in telephony transmit two-state digitally encoded information in the form of a serial stream of pulses. Transmission systems can be divided into two major categories and many subcategories. The major categories chosen for discussion are <u>frequency division</u> and <u>time division</u> systems. In frequency division systems, channels are separated by frequency assignment. In time division systems, channels are separated by time assignment.

1.4.1 Frequency division systems for voice communications are generally carrier modulated on a continuously changing basis (noninterrupted), and referred to as <u>analog</u> carrier. Analog and frequency division have come to mean the same in carrier systems. Examples of analog, frequency division carrier are amplitude modulation (AM) and frequency modulation (FM) systems. In most cases, FM systems are in reality phase modulation (PM) systems, but are referred to as FM. The most common analog carrier systems used in telephony are double sideband (DSB) AM, single sideband (SSB) AM, and FM systems. Most DSB AM systems transmit a carrier signal for each channel and most SSB AM systems suppress the channel carrier, and transmit one or more pilot signals for regulation of channel groups or systems.

1.4.2 Time division systems sample analog signals and transmit information representing the instantaneous value of that signal at the time it was sampled. Time division systems can be subdivided into two major groups. The first major group utilizes modulation techniques such as pulse amplitude modulation (PAM), pulse width modulation (PWM), pulse postion modulation (PPM) and similar systems. These systems transmit the modulated signal in a form that degradation can occur on a gradual and cumulative basis. The other major group of time division systems is called digital systems. In digital systems, information is transmitted as serial stream in a state such as: Pulse or no pulse; on or off; or one or zero. Gradual or cumulative degradation is unlikely in digital transmission systems because the information is periodically regenerated rather than amplified. Digital systems generally work well to a crash point and then fail completely.

1.4.3 Pulse Code Modulation (PCM) is the most widely used digital system in telephony. Each voice sample level is encoded into 8 serial bits of information and transmitted as an 8-bit code (D3 and D4 types). Digital subscriber carrier, digital concentrators, and digital central offices use D3 voice encoding, but signaling information is not standardized for subscriber equipment and services. Another form of digital system is called differential PCM (DPCM) or delta modulation. DPCM systems transmit changes from the last sampled state. Most digital systems now use T1 span lines as the transmission medium (1.544 megabits per second).

1.5 PCM Fundamentals: Perhaps the best way to illustrate the fundamentals of digital transmission systems is to follow the encoding, transmission and decoding process of a pulse code modulation (PCM) carrier system. Appendix A is a brief illustrated guide on the D3-T1 trunk carrier system. PCM subscriber carrier and other digital systems in telephony operate in a similar manner.

2. HIERARCHY

2.1 A basic digital transmission hierarchy was established by the Bell System in the early 1960's. The basic building block was 24 voice channels of PCM carrier combined to form a 1.544 Mb/s line signal.

Designation	Rate (Mb/s)	Voice Channels
T1	1.544	24 (T1)
T2	6.312	96 (4x T1)
T3	44.736	672 (7x T2)
T4	274.176	4032 (6x T3)

2.2 Each digital voice channel is sampled 8000 times per second times 8 bits per channel times 24 channels, requiring 1.536 Mb/s of channel information for each 24 channel system. To this is added a framing bit for each of the 8000 samples, resulting in a T1 bit of 1.544 Mb/s. The 96 channels of a T2 system may originate in as many as four different locations of 24 channels each over separate T1 lines. Each of the T1 systems may operate from independent 1.544 Mb/s clocks. The slight differences in bit rates result in asynchronous (not synchronized) signals being fed into a multiplexer. The T2 rate of 6.312 Mb/s has 136,000 bits over that of four

times the T1 rate (6.176 Mb/s). The multiplexer utilizes these extra 136,000 bits to maintain a 6.312 Mb/s rate from the four random 1.544 Mb/s inputs. This is done by inserting control, framing and pulse stuffing bits. Stuffing bits are added as required to synchronize the four random bit streams. These extra bits are discarded when the signal is demultiplexed. Other special processing also takes place in multiplexers to improve the signal format for transmission (i.e., suppression of a long string of zeros or no pulse condition).

2.3 The original conceptual hierarchy has been further refined to address present day needs and hardware. The D channel banks have been expanded, reflecting improvements in technology. The T transmission systems still follow the basic format, but have been expanded to include intermediate multiples of the basic channel capacities. These changes primarily reflect changes in technology, economics and resultant hardware. The T designation basically identifies Western Electric equipment (hardware). The Independent manufacturers have developed functionally equivalent digital systems, and the D and T designations often imply equipment compatibility standards. While D and T designate equipment, the terms DS and DSX were implemented to designate system bit rates and interface standards. DS1 refers to a digital signal at the first level, or 1.544 Mb/s. DSX1 is a crossconnect or interface point for a DS1 signal. This is the present hierarchy of digital signals.

Designation	Rate (Mb/s)	Voice Channels
DS1	1.544	24 (T1)
DS1C	3.152	48 (2x T1)
DS2	6.312	96 (4x T1)
DS3	44.736	672 (7x T2)
DS4	274.176	4032 (6x T3)

2.4 The digital hierarchy with standard multiplexers and interface points are illustrated in Figure 1. Intermediate levels such as DS1C may be converted back into the next lower level before entering the hierarchy at higher levels. The voice channel capacity is shown above; in reality, the bits may be used for voice, data or combinations of voice and data. The Independent manufacturers adhere to the Bell System hierarchy, but have added variations to this hierarchy to address specific market and application situations of the Independent telephone companies such as radio and optical fiber systems. This will be further discussed.

3. IMPLEMENTATION ON PAIRED CABLES

3.1 This phase of the implementation of digital systems addresses hardware applied to exchange grade cables and other special types of paired cables. This phase is considered a beginning level and generally deals with digital systems at the exchange level. Digital systems applied to optical fibers, coaxial cables and radio will also be discussed in later paragraphs. These are generally considered to be higher order systems (high bit rates) utilized in the intertoll network. However, the practical applications for fiber systems may extend down to DS1 and DS2 rates, and may find large scale application at the exchange level in the future.

3.2 T1 carrier was designed for use in metropolitian areas. The developed technology and hardware have expanded the application of this equipment

such that few areas of telephony remain untouched by digital transmission and switching. In rural telephony, the vast majority of new trunks are supplied by T1 carrier systems. This trend became evident in 1968, the first year that REA began evaluating and listing PCM carrier developed by Independent manufacturers. Of the trunk carrier channels supplied to REA borrowers under contract that year, 30 percent were PCM carrier.

3.3 Bell Labs established T1 span line engineering guidelines in 1963. Maximum repeater spacings were established based on extensive cable data analysis. As cables were loaded with T1 systems shorter spacings were required to offset the effects of near end crosstalk. Superior Cable Corporation developed a T Screen cable for effectively separating the transmit and receive cable pairs in 1970. This provided the effect of separate cables for transmit and receive directions, and provided for full repeaters spacings. Filled core cables became widely used in the early 1970's. The lower attenuation characteristics of filled cables provided for 10 to 15 percent longer repeater spacings. Improvements in cable screen techniques provided additional near end crosstalk isolation, allowing for the application of higher bit rate digital systems.

3.4 The earlier D1 channel banks were utilized at the exchange level for toll connecting and EAS service. Later channel banks were designated D2, D3 and D4. The significant technical difference is that D1 uses seven bits to encode each voice sample, resulting in 127 encoding steps; the D2, D3 and D4 use eight bits to encode each voice sample, resulting in 255 encoding steps. The result is less channel noise and distortion with the later D channel banks. The D4 channel bank (or Independent manufacturer's equivalent) is emerging as an economical and flexible channel bank that will interface with other D3 or D4 channel banks in synchronized or non-synchronized modes of operation. The D4 output can be arranged to directly interface T1, T1C and T2 span lines. The 1,544 Mb/s bit stream can be readily accessed on a per channel basis for data (in 64 kb/s increments).

3.5 Variations of the D1, D2, D3 and D4 trunk channel banks have been adapted for subscriber service by Independent manufacturers. These systems have provided subscriber service almost exclusively over exchange cable pairs utilizing T1 span lines. Present subscriber carrier utilize D3 voice encoding; but each manufacturer differs slightly in the use of signaling bits to accomplish dialing, ringing, etc. Also, there are variations in alarms, housekeeping and remote testing functions between manufacturers. While most manufacturers have used only signaling bits to transmit this information between the office and subscriber terminals, some use part of the framing (synchronization) bits also. Subscriber line concentrators are integrated into some types of digital subscriber carrier.

3.6 Bell Labs developed a special paired cable for T2 transmission in 1970. This second step in the digital hierarchy has been essentially dormant, especially in the Independent telephone industry. The benefits versus costs of T2 applied to special paired cables are marginal at best.

3.7 Responding to needs within the capability of existing cable plant and technology, variations of the original hierarchy began to be used. An example is T1C with 48 voice channels transmitted at 3.152 Mb/s. Exchange cables may be inadequate or marginal for T1C application. This is especially so for subscriber cables where cables are frequently tapped

and where housekeeping may suffer. Independent manufacturers have addressed this market with T1C equivalent systems, and with variations of T1C that compress the 3.152 Mb/s information rate into span line signals at lower rates and impose significantly lower transmission requirements on cables than T1C. Compressed bit rate systems are primarily intended for application to existing exchange cables that will not support higher bit rate systems.

4. IMPLEMENTATION ON IMPROVED FACILITIES

4.1 Higher order digital systems (above the T2 rate) cannot economically be transmitted over paired cables. In concept, the alternative transmission media for these higher order digital systems was coaxial cables, waveguides and radio. Optical fibers began to show significant promise as a digital transmission medium in the mid 1970's.

4.2 Lightwave systems in service are generally at T3 channel capacities or multiples of T3. Lightwave systems are not yet fully standardized. Current lightwave systems with T3 channel capacities may utilize bit rates slightly higher than DS3 (44.736 Mb/s). These extra bits are used for alarms, housekeeping and synchronization functions of the lightwave systems. These systems generally are not end to end compatible on an optical basis, but can be made compatible on an electrical interface basis at standard DS levels.

4.3 There is also current work in progress to investigate the economic and technical feasibility of lightwave systems using bit rates substantially lower and higher than T3. For lower density applications, the economical use of T2 and even T1 rates over fibers is being reviewed. It would be premature to judge the outcome of this work. Higher bit rates such as two times T3 (1344 channels) may become an intermediate standard for fiber systems. The use of longer wavelength systems provides for lower fiber attenuation and few repeaters between terminal locations.

4.4 Analog and digital radio systems are also being used for the transmission of digital signals. Standard T1 signals (1.544 Mb/s) or multiples of T1 are processed and multiplexed onto the radio. The digital system may use external multiplexers for application to analog radio, or the digital signals may be directly applied to digital radio. A digital radio is the functional equivalent of analog radio with an external digital multiplexer. Most digital signals applied to radio in the Independent industry do not follow the standardized Bell System digital hierarchy and format.

4.5 Lower density digital radio transmission systems are available for rural applications. The channel capacities of these systems are 24 (T1), 48 (2 x T1), 96 (4 x T1), 144 (6 x T1), 192 (8 x T1) and higher. Four or eight level phase-shift keyed (PSK) modulation is commonly used to achieve bandwidth efficiency. Digital signals such as T1 signals generally contain repetitive patterns, causing high amplitude discrete frequency components. Before these signals are modulated on the radio, they are scrambled (encoded) to eliminate any repetitive patterns. This reduces the radio loading.

4.6 The digital radio systems of one manufacturer are generally not end-to-end compatible with equipment of other manufacturers at this time. However, these systems can be standardized at the DS1 level (or higher) for transmission over T1 span lines. Thus, they can serve as a digital transmission building block along with other standard systems in the digital hierarchy.

4.7 The original digital hierarchy planned the use of T4 systems over coaxial cables. This was to be a long haul system carrying 4032 voice channels (168 times T1). In rural areas, the need for 4000 voice channels in one route is non existent at this time or in the foreseeable future. Where a need for large quantities of voice channels does exist, lightwave systems are likely to be the economical choice.

4.8 In rural areas, coaxial cables may be used to provide CATV or other broadband services. Voice channels and data may also be applied to small bandwidth segments of these coaxial cables to provide both broadband and narrowband services. Digital techniques are especially efficient for data services and T1 carrier systems are very cost effective for voice circuits. Utilizing multiplexers and techniques very similar to digital systems applied to analog radio, segments of coaxial cable systems can serve as building blocks in the digital hierarchy.

5. DIGITAL SYSTEM MODULES

5.1 Digital transmission and switching systems are designed and assembled in modules. Each module is essentially separate and complete as a subsystem, but can be integrated to form systems in a digital network.

5.2 Each digital module is designed to some interface standard (not necessarily an industry wide standard) such that it can be considered a separate and complete unit for installation, alignment and testing. The architecture for the various digital transmission and switching systems are designed to meet certain industry requirements for direct digital interface, but there is no universal agreement on architecture and interface standards.

5.3 An integrated digital network can be formed with currently available hardware. Figure 2 shows some of the digital transmission systems available for the trunk network; and Figure 3 shows some of the digital building blocks for subscriber service. All of these digital modules may not be available from one manufacturer; and there is a high probability that equipment from different manufacturers will only interface on an analog voice frequency basis. This is rather universally true for subscriber service. Nonstandard interfaces exist with some trunk systems, but these systems will interface digitally at DS1 and higher bit rates in the Bell System defined hierarchy. To achieve greater efficiencies, manufactuers have combined DS1 signals into higher bit rate systems for application to paired cables, radio, coaxial cables and optical fibers. While standardization at all bit rates would be desirable, it may not be the most cost effective use of the transmission media. Where nonstandard interfaces are used for transmission, they can be converted into standard DS levels at interface points.

5.4 <u>Trunk Systems</u>: The following are examples of digital systems that can be used to form a digital trunk network.

5.4.1 Digital COE: The digital central office equipment is rather universally defined for trunk service. The standard interface is DS1 (1.544 Mb/s) and the decoded digital signal is equivalent to a D3 channel bank. In practical terms, the digital COE can be used to interface with other digital COE and D3 channel banks for almost all switched services. It has not been economically practical to provide non switched full period circuits (nailed up circuits) for special services. A separate D3 channel bank is currently used for this service -- even if only one full period circuit is required. (Note: There are techniques for intercepting the DS1 bit stream to extract one of the 24 channels before it enters the digital switch. Such systems must be considered experimental at this time.)

5.4.2 D3 Channel Bank: The D3 (or D4) channel bank (24 channels at the DS1 rate) is the universal standard for trunks in the digital network in North America. Digital central office equipment is designed on the basis of D3-T1 interface.

5.4.3 T1 Span Line: The T1 type span line (DS1 rate) on paired telephone cable is the standard interface for systems in the digital network. Higher bit rate systems are also used. These higher bit rate systems may not be standard, but can be interfaced at the DS1 rate.

5.4.4 Digital Radio: Digital radio (or analog radio with digital multiplexer) can be used economically for point-to-point digital transmission at multiples of the DS1 rate. Most digital signals applied to digital radios above the DS1 rate do not follow the standardized Bell System digital hierarchy and format. Low density digital radios with voice channel capacities up to 192 (8 x T1) are available for rural applications. Higher density digital radios providing up to 1344 voice channels (56 x T1) are also available.

5.4.5 A-D Converter: An analog-to-digital converter (A-D converter) can be an economical technique for some telcos. An A-D converter transforms the voice and signaling of an analog carrier system into a digital format (and a digital bit stream into an analog format). The only A-D converters available today are for L carrier to T carrier. The unit is designed to convert two 12 channel L carrier groups (voice and signaling) into one 24 channel D3-T1 digital signal (DS1). The primary application is where L carrier exists (over radio or coaxial cable) and there is a need to interface a digital COE, or extend the circuits over digital facilities such as T1 span lines.

5.4.6 Lightwave Systems: Lightwave transmission systems may be cost effective for high density digital applications; the feasibility of low density fiber systems is also under study. Standardization on a lightwave basis does not exist today between different manufacturer's fiber systems. These systems can be interfaced on an electrical basis at standard DS rates.

5.4.7 Coaxial Cable Systems: Digital transmission over coaxial cables will not generally compete with other techniques. They may become economical in the future when combined with other systems such as CATV.

5.5 Subscriber Systems: Digital compatibility of digital subscriber systems exist only to a minor extent. At this time, standards exist only for

T1 type span lines and for D3 voice encoding. Equipment from different manufacturers are unlikely to interface on a digital basis. The following are examples of digital building blocks for subscriber service.

5.5.1 Digital COE: Only the D3 voice encoding and DS1 interface are defined for digital central office equipment in subscriber service. The signaling, processing, alarms, housekeeping and remote testing functions are not standardized among manufacturers. In general, equipment from different manufacturers can only be interfaced on an analog voice frequency basis. A typical remote switching terminal (RST) handles 250 to 350 lines, and is connected to the host COE on a direct digital basis over two T1 span lines. The digital COE architecture varies with manufacturer and equipment. These hardware and software differences do not allow for easy mixing of equipment. There are several alternatives offered for serving small groups of subscribers on a direct digital interface basis. One method is to distribute the 250 to 350 lines at multiple RST locations. Another method is to utilize subscriber terminals of PCM subscriber carrier and interface the COE on a direct digital basis (omit CO channel banks). The PCM subscriber carrier may interface the RST as well as the COE.

5.5.2 Digital Subscriber Carrier: PCM subscriber carrier designed in 24 channel groups with D3 voice encoding and T1 span lines is in wide use on an analog interface basis. Some types are designed to interface the digital COE or RST on a direct digital basis (if all equipment is furnished by the same manufacturer). Lacking standardization in signaling, alarms and testing functions, its role on a direct digital interface basis may be limited. However, digital subscriber carrier is expected to maintain a strong role in subscriber service on an analog interface basis because of its simplicity, versatility, and low cost.

5.5.3 Digital Subscriber Line Concentrators: Digital concentrators combine digital transmission and digital switching to interface the COE on an analog basis. Some "concentrators" are designed to interface the COE on a direct digital basis (when furnished by the same manufacturer). By definition, it then ceases to be a concentrator and becomes an RST. Digital concentrators are gaining in popularity as an alternative to RST's and subscriber carrier for larger circuit quantities at one location.

5.5.4 T1 Span Line: The T1 span line (DS1 rate) on paired telephone cable is the standard interface for digital systems in subscriber service. Higher bit rate systems are available, but may have technical and practical limitations in subscriber service. Higher bit rate span lines (i.e., 2 x T1) may offer more circuits per cable pair; but many subscriber systems require two T1 span lines and the two separate paths are desired for reliability. It would serve little purpose to increase the span line bit rate if additional alternate paths are required for reliability.

5.5.5 Digital Radio: Digital radio in subscriber service is almost identical to the trunk application. Low density digital radio can be used to serve pockets of subscribers. Digital radio can be used in conjunction with other digital hardware such as subscriber carrier, concentrators and RST's. The digital radio signal can easily be separated into DS1 signals and extended over T1 span lines.

5.6 Miscellaneous Digital Equipment: Two other types of digital equipment to be discussed are automatic protection switches and maintenance and

test systems.

5.6.1 Automatic Protection Switches: Automatic protection switches (APS) were developed initially for high density analog radio systems to achieve very high service reliability. The inherent nature of digital systems provides for low cost APS systems to protect small channel quantities. Digital APS systems are available for paired cable systems, lightwave systems and radio systems. They are arranged to protect in levels or groups, beginning at the basic DS1 bit rate. The digital bit stream is transferred to an alternate path when the main path becomes unusable. Error detection and transfer is fast, and circuits maintain service continuity. REA Specification PE-60c covers a universal APS for T1 span lines. Automatic transfer and restoral is required. There are systems in use that are not universal (require special interface equipment) and that do not have features such as fast speed and automatic restoral. These systems are limited in capability and are not generally recommended for use.

5.6.2 Maintenance Systems: Digital techniques offer enhanced possibilities for manual and automated system maintenance, alarms and testing. The initial T1 span lines were designed with the capability to interrogate repeaters from remote locations. Status reporting, housekeeping functions, alarms, manual and automatic testing of remote terminals, and other features may be integrated into the equipment, or available as separate equipment. Advances in microprocessors have made these features more versatile and more economical. Available equipment ranges from simple, low cost units to complex, costly systems.

5.7 Future: Standardization has been a primary key to the success of digital transmission systems applied to paired cables. Standardization is limited in integrated digital transmission and switching systems, and for the emerging technologies such as lightwave systems. Premature standardization could severely slow the rapid development pace of these emerging systems. This should not be a major concern so long as partial standardization provides for the economical use of these systems. Such is the case for optical fiber systems which are not standardized on an optical basis, but can be electrically interfaced at standard DS levels. However, the economics expected from integrated digital subscriber systems may not fully materialize because of the lack of standardization. It is for this reason that digital subscriber carrier, digital concentrators and other non integrated digital subscriber equipment will continue to maintain a strong role in subscriber service.

FIGURE 1

STANDARD DIGITAL HIERARCHY

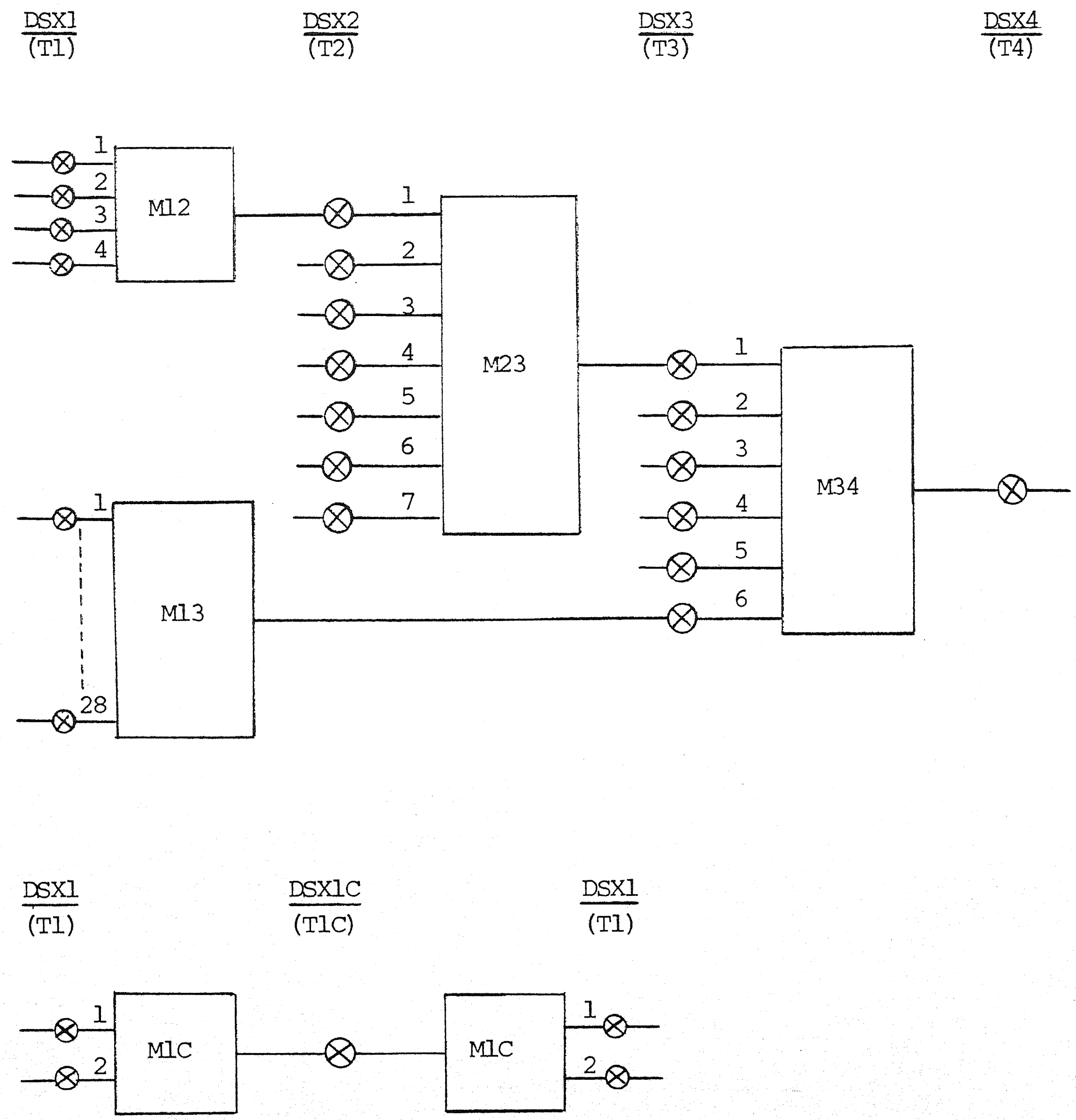
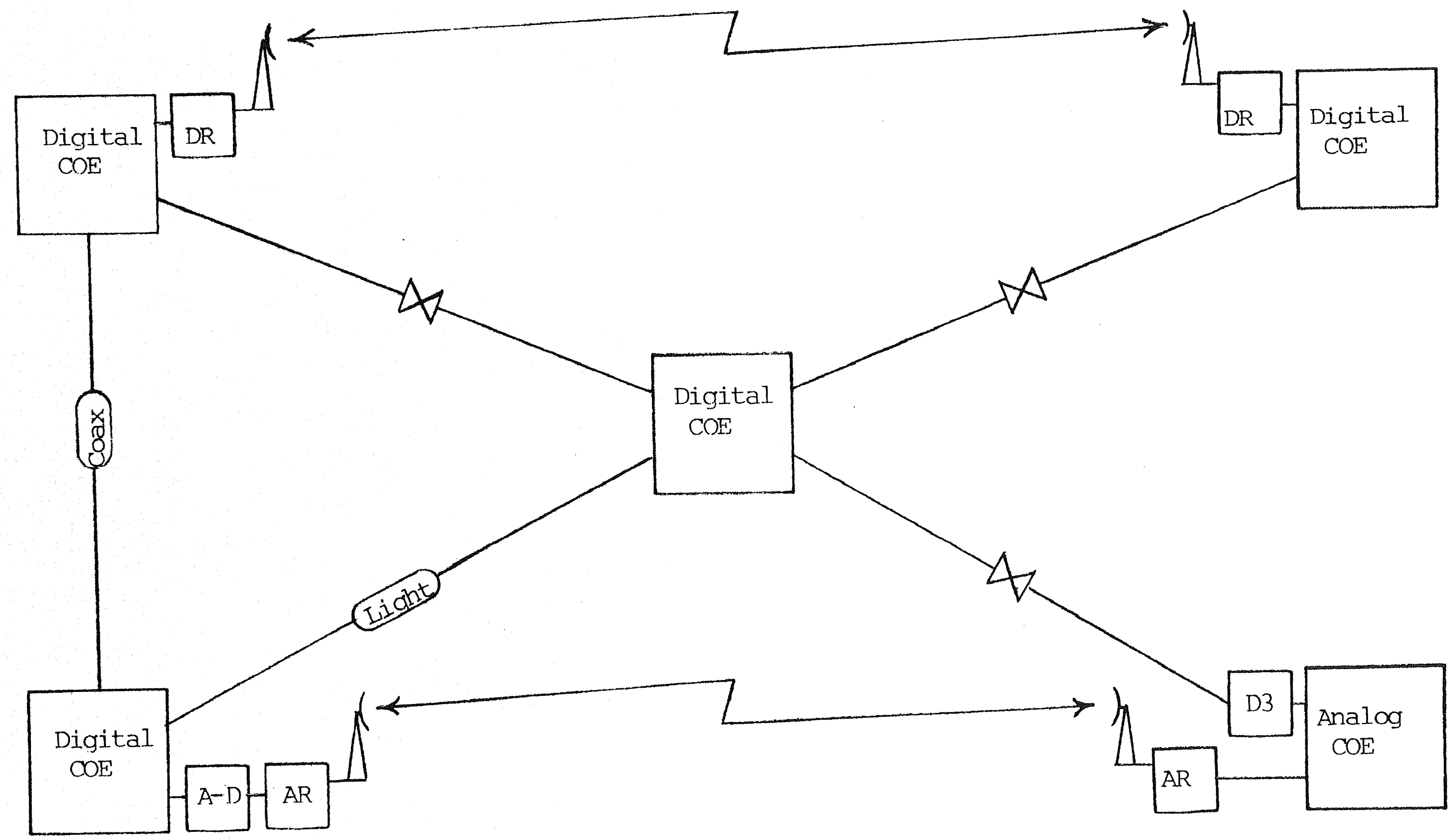


FIGURE 2

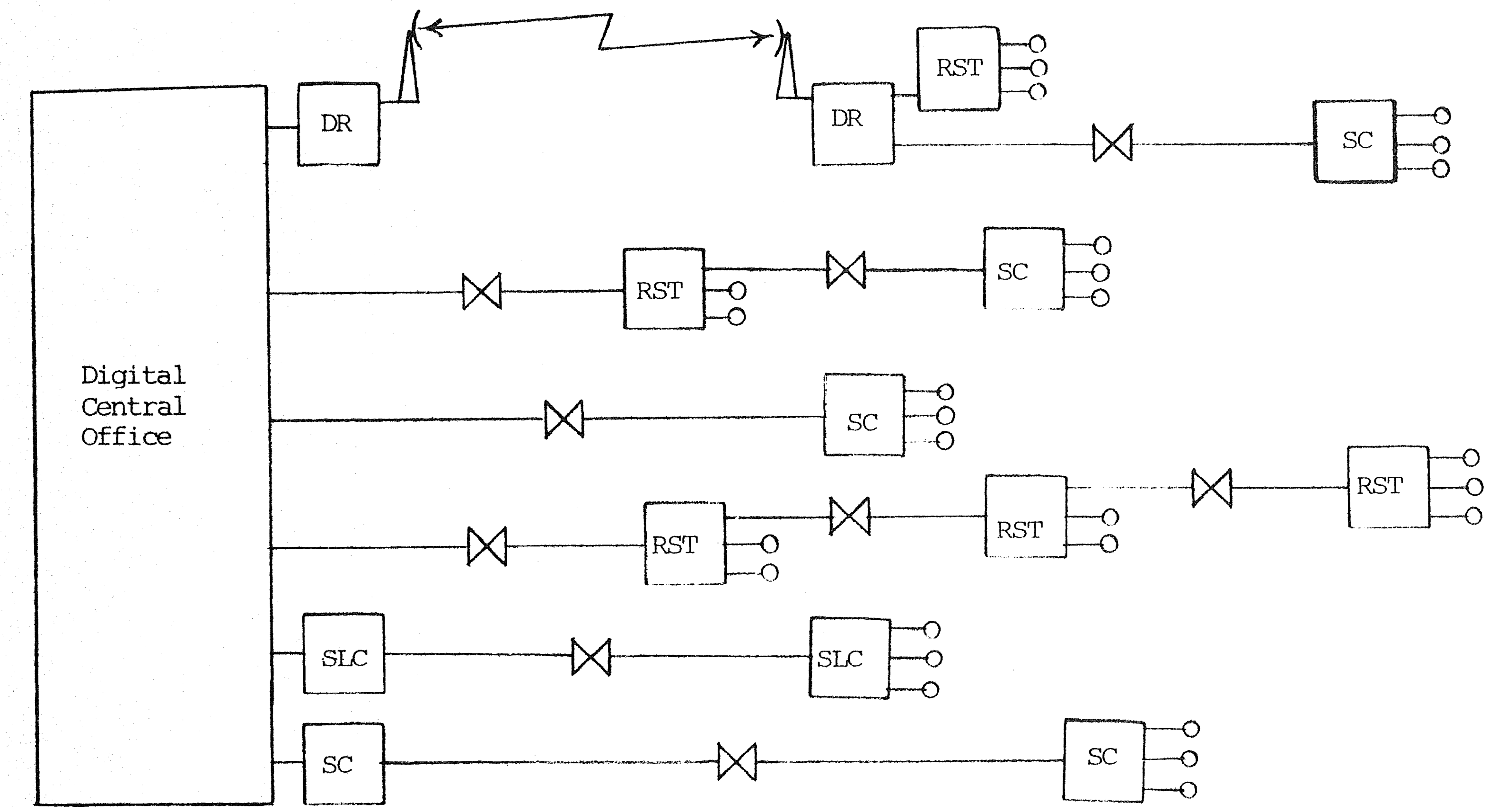
DIGITAL TRANSMISSION SYSTEMS FOR TRUNKS



NOTES: ⋈ = Digital Span Line (T1 or Higher on Paired Cables)
D3 = Digital Carrier Channel Bank (D3 or D4)
DR = Digital Radio
A-D = Analog to Digital Converter (L Carrier to T Carrier)
AR = Analog Radio (L Carrier Multiplex)
(Light) = Lightwave Digital Transmission System
(Coax) = Coaxial Cable Digital Transmission System

FIGURE 3

DIGITAL BUILDING BLOCKS FOR SUBSCRIBER SERVICE



NOTES: —O = Subscriber
RST = Digital Remote Switching Terminal
SC = Digital Subscriber Carrier
SLC = Digital Subscriber Line Concentrator
DR = Digital Radio
⋈ = Digital Span Line (T1)

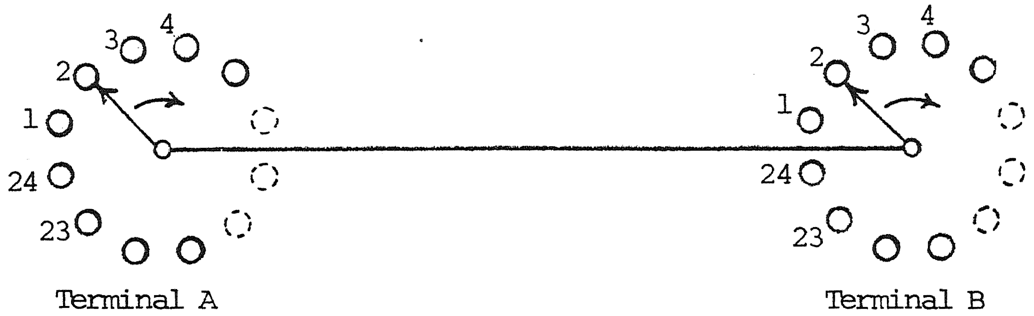
SC and SLC can interface COE on an analog (voice frequency) basis, or in some cases on a digital basis where a matching interface is available.

APPENDIX A

PCM CARRIER FUNDAMENTALS

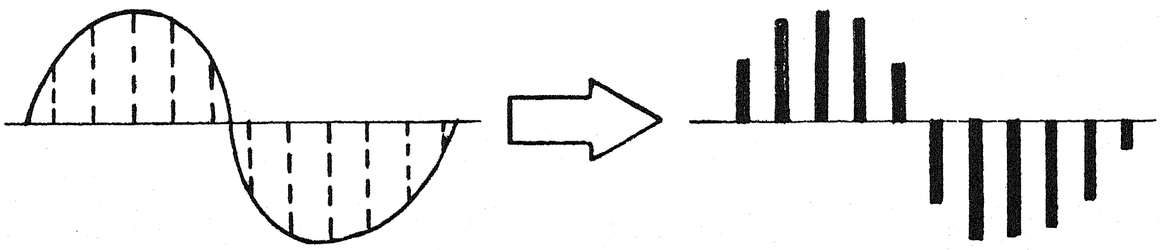
A pulse code modulation (PCM) carrier system converts the continuously changing (analog) voice signals into coded pulses at a periodic sampling rate. These pulses are transmitted over cable pairs, reconstructed when they become weak, and decoded at the distant end. The D3 PCM terminal and T1 span line will be used to illustrate this process, beginning with time division multiplexing.

Time Division Multiplex



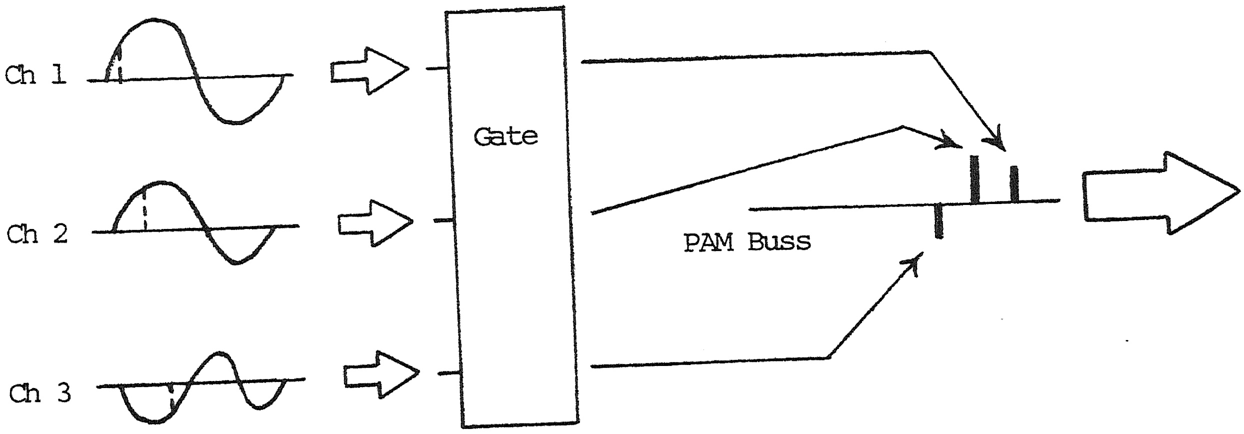
A D3 PCM carrier system has 24 channels or voice paths. Switches or gates at both terminal ends must be synchronized so that channel 1 samples of Terminal A are received by channel 1 of Terminal B. Each channel is sampled at 8,000 times per second which is controlled by a precise clock.

A voice signal is sampled and converted into discrete voltage levels as shown.



This is called pulse amplitude modulation (PAM).

These PAM signals are fed into a PAM bus from each channel in sequence.



Each PAM signal is converted into "bits" of encoded information. Each group of bits represents a specific PAM voltage level. This is called binary coding.

Binary Coding

Binary refers to two possible states —

ON or OFF   1 or 0   PULSE or NO PULSE

Binary is a simple numbering system using a base number of two. Binary coding is easily adapted to semiconductor technology, and is an efficient method of storing and transmitting information. By using ones and zeros, any number can be stored and transmitted.

With 1 digit or store, we can count 2 numbers
2 digits: 2 x 2 = 4 numbers
3 digits: 2 x 2 x 2 = 8 numbers
4 digits: 2 x 2 x 2 x 2 = 16 numbers
The maximum number doubles with each digit.

Binary coding is used in PCM carrier systems. A "1" represents a pulse or "on" condition, and a "0" represents a no pulse or an "off" condition. A 3 bit code can be used to represent 8 information levels, or to count from 0 to 7. (Zero is an information level or number.)

Number	Binary Code
0	000
1	001
2	010
3	011
4	100
5	101
6	110
7	111

This is another example of a 3 bit code with + and - levels or numbers.

Number	Binary Code	
0	000	OMIT
-1	001	
-2	010	
-3	011	
0	100	
+1	101	
+2	110	
+3	111	

The first bit defines polarity; + is 1 and - is 0. The next two bits count from zero to 3. Only one code is needed for zero. Generally the all zero code is omitted so that pulses will be available to drive the system clocks.

There are 8 possible levels for a 3 bit code. Each added bit doubles the possible code levels. There are 8 bits in the D3 encoding used today. This allows for 256 possible levels. The D3 system uses 127 "+" levels, 127 "-" levels, and zero, or 255 levels or codes.

Each PAM signal is converted into 8 bits of encoded information for each channel. For example:

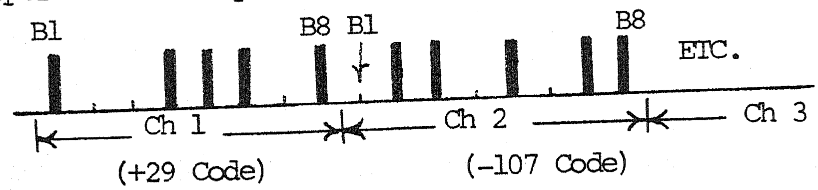
Channel	Bit	1	2	3	4	5	6	7	8
1	Code	1	0	0	1	1	1	0	1
2		0	1	1	0	1	0	1	1
3		ETC.							

Bit	Ch 1	Ch 2
1	+	-
2	0	64
3	0	32
4	16	0
5	8	8
6	4	0
7	0	2
8	1	1
Value	+29	-107

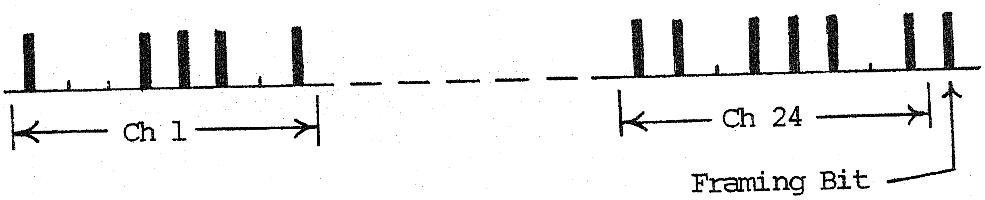
Bit	1	2	3	4	5	6	7	8
Weight	±	64	32	16	8	4	2	1

A "1" represents a pulse and a "0" represents a no pulse condition. (This is further discussed under "Encoding".)

The bits appear on the PCM buss as unipolar pulses (in groups of 8) in proper channel sequence.



The sampling of 24 channels is called a frame. An extra information bit is added at the end of each frame, and is called the framing bit.

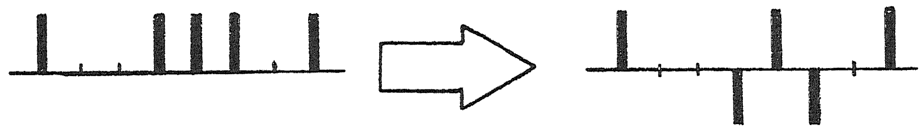


There are 8 bits per channel times 24 channels equals 192 bits, plus one framing bit equals 193 bits per frame.

193 bits per frame times 8,000 samples per second equals 1,544,000 bits per second.

The unipolar pulses are then converted into bipolar pulses, alternating in polarity (alternate bipolar pulses).

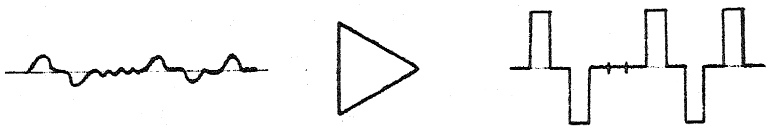
Unipolar Bipolar



Bipolar pulse transmission has advantages over unipolar pulse transmission.

1. Power is concentrated near 772 kHz.
2. Low frequency power is reduced (no dc).
3. High frequency power is reduced.
4. Errors can easily be recognized.

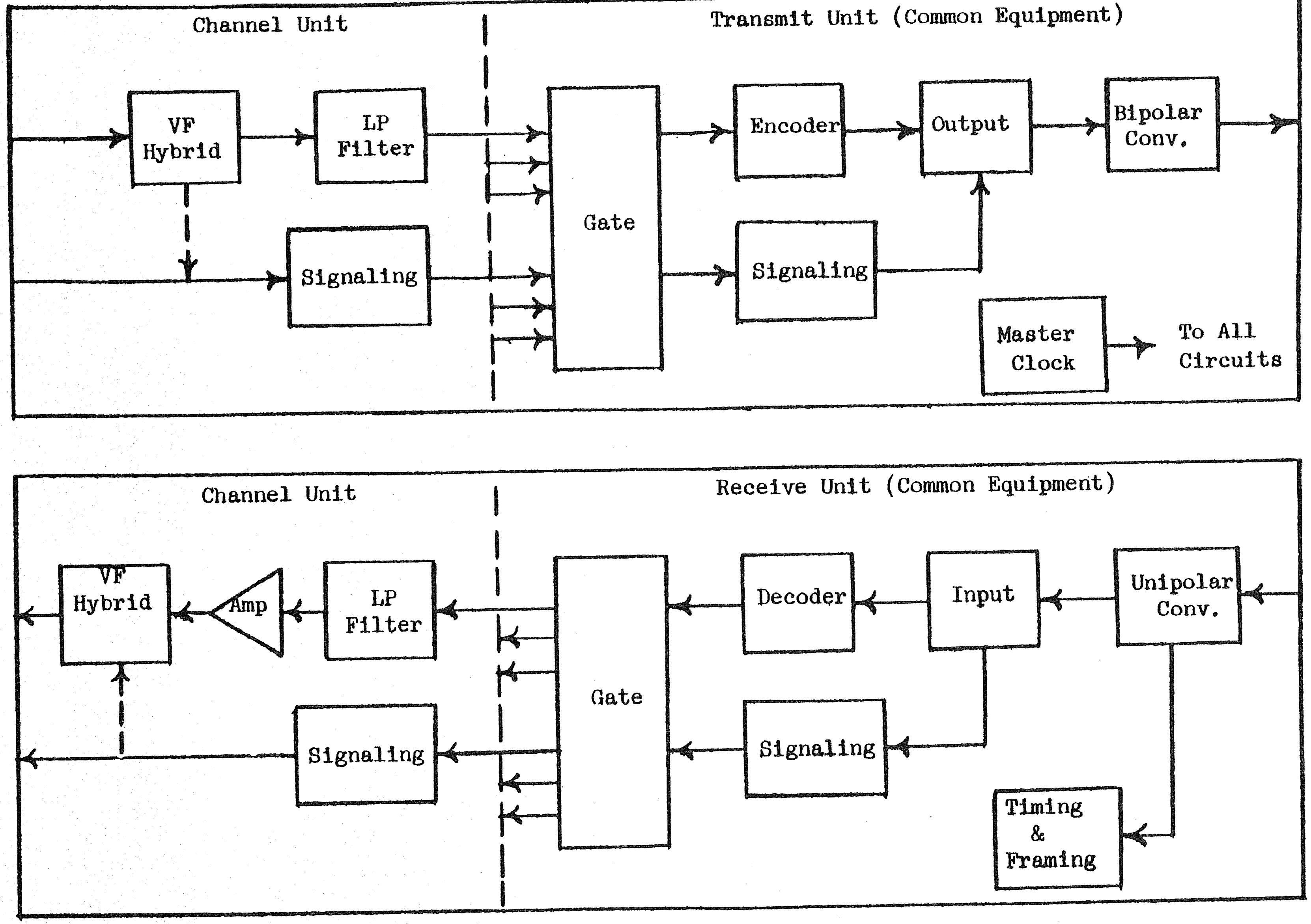
As the pulses are transmitted along the cable, they become weak and distorted. The repeater will sample the incoming signal 1,544,000 times per second to determine if there is a pulse or no pulse condition. If there is a pulse present at the input, a new pulse of the same polarity will be generated at the output. The sampling rate is controlled by a clock (tuned circuit) and the decision level is controlled by a threshold detector within each repeater.



Input Repeater Output

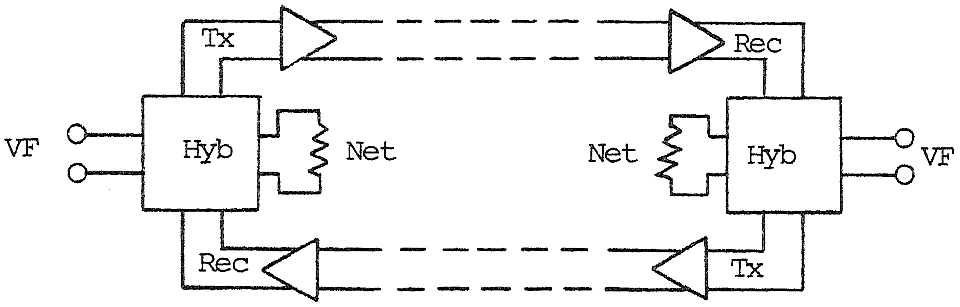
At the distant terminal, the 8 bit coded groups are converted back into PAM signals, distributed to the proper channels, and filtered (low pass filter) to restore the analog voice signals. Some distortion is added to the voice signal. This distortion is a function of the terminal equipment basic design. Unlike analog systems, noise and distortion are not added along the transmission path.

A simplified sketch of a PCM transmitter and receiver is shown on the next page.



PCM CARRIER TRANSMITTER AND RECEIVER

A PCM carrier system is a two-way four-wire transmission system. The functions are duplicated at both ends. A voice frequency 4 wire termination set is used to combine the carrier transmit and receive signals at the cable pair (2 wire).



Framing

The framing bits are used to synchronize the transmitter and receiver. The framing bits are generated in the transmitter and separated in the receiver for proper timing and framing. These framing bits are in a special sequence of pulses and spaces for 12 frames. The 12 frames are called a super frame.

1	2	3	4	5	6	7	8	9	10	11	12	Frame
1	0	0	0	1	1	0	1	1	1	0	0	Framing Bits
					A						B	Signaling Channel

Signaling

One of the 8 voice bits (B8 - the least important) is used for signaling each 6th frame. The PCM voice encoding is actually 7 5/6 bit encoding, or almost 8 bit encoding.

There are two signaling channels. Each voice channel transmits one bit of signaling during the 6th frame (signaling channel A) and another signaling bit during the 12th frame (signaling channel B). Each signaling channel contains:

$$8000 \div 6 = 1333 \text{ bits per second.}$$

The signaling channels transmit all of the dialing, ringing, and supervisory information from one terminal to the other.

(Note: The voice encoding, framing and signaling foremat are well defined for a D3 PCM trunk carrier system. There are variations in the framing and signaling foremat for PCM subscriber carrier, PCM carrier-concentrators and digital class 5 central offices of different manufacturers.)

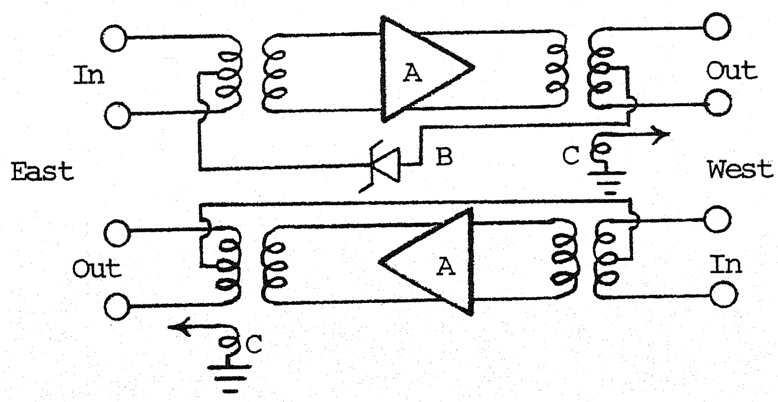
Encoding

PCM carrier uses nonlinear encoding. With earlier D3 terminals, this is accomplished by a compandor (compressor and expandor) and a linear encoder and decoder. Later D3 (or D4) terminals utilize a nonlinear encoder and decoder. The same nonlinear encoding results from either method. Much improvement in voice quality with fewer coding steps results from this nonlinear encoding. The coding steps are small for low level samples and larger for high level samples. The coding steps are determined by a companding law of $\mu = 255$.

There are two possible states for each "bit" of information. They are expressed as pulse or no pulse, on or off, one or zero. Two bits have 4 possible combinations or levels, 3 bits have 8 possible levels, and 8 bits have 256 possible levels. This is called binary coding. Codes represent 127 positive levels, 127 negative levels and zero. This utilizes 255 of the 256 possible codes; the all zero code is not used. Of the 8 bit group, the first (B1) represents polarity; the second (B2) represents the largest value and the last (B8) represents the smallest value. Thus, B1 is the most important bit and B8 is the least important. Each 8 bit code can be converted into a positive or negative voltage sample of a specific value.

Repeater

A PCM repeater contains two regeneraters (A), a power supply (B), interregation circuits (C), and electrical protection (shown later).



Regenerator

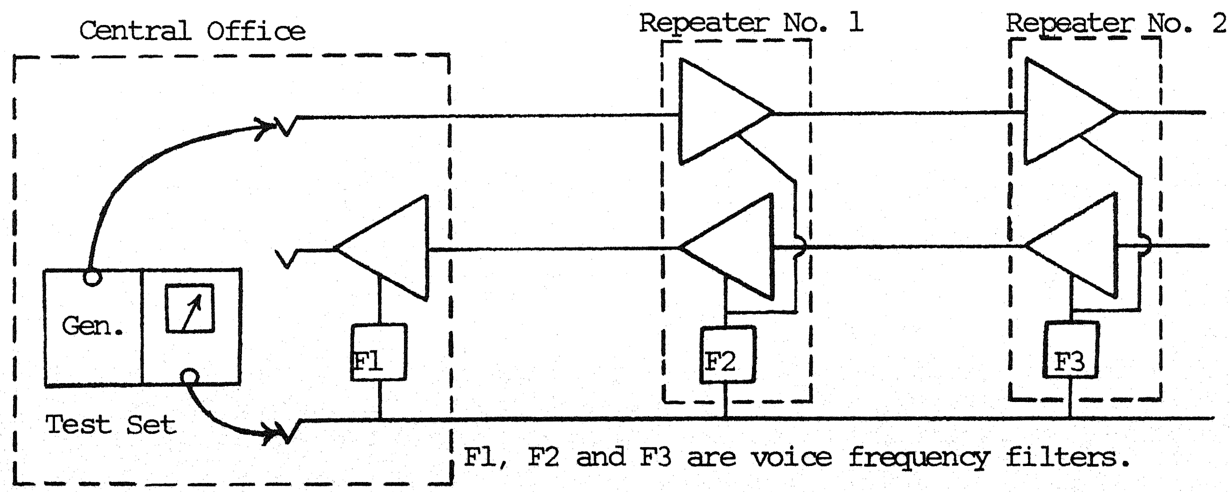
The regenerator samples the input at precise intervals and regenerates new pulses of the same polarity at the output as described earlier. The PCM span line signal alternates the polarity of each generated pulse. Two pulses of the same polarity indicate that an error has occurred. A bridging error detector used at the input and output of each repeater can determine the faulty repeater or cable section causing the errors. An acceptable error level is about one out of one million bits. This varies with the specific service and application.

Repeater Power

The span power is fed down the PCM carrier transmit cable pair and returns on the carrier receive cable pair. As the current passes through the repeater, a zener diode causes a constant voltage drop of about 10 volts. This voltage is the power supply for the repeater. A capacitor is added across the zener to improve the immunity to 60 hertz current in the cable pair.

Repeater Interrogation

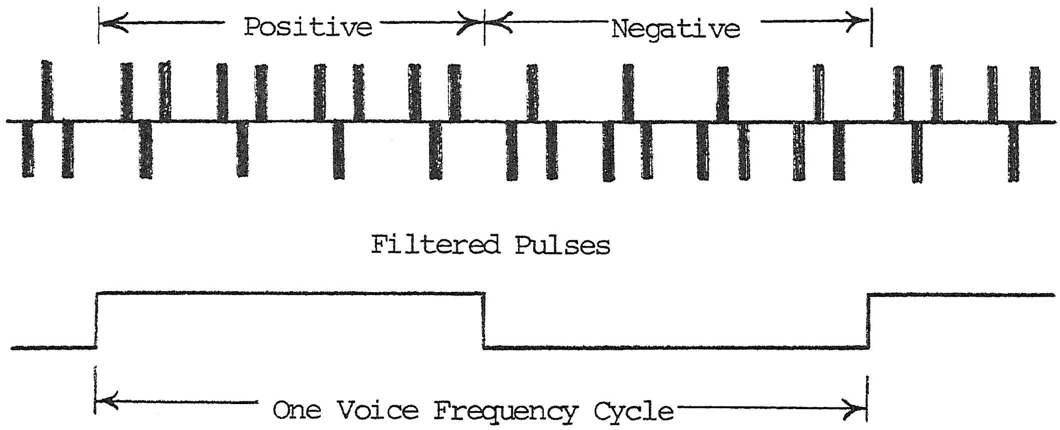
PCM repeaters contain interrogation circuits. A digital bit stream containing specific error patterns is generated at the central office and transmitted down the span line. These error patterns are detected in the third winding of the repeater output transformer and passed through a voice frequency filter. There are 12 filters associated with the error patterns. Each repeater location uses a different voice frequency filter. In most cases of span line troubles, the faulty repeater can be determined from the central office.



F1, F2 and F3 are voice frequency filters.

The interrogation test set consists of a pulse generator and a voice frequency selective voltmeter. The pulse generator output is connected to the transmit side of the span line, and provides the span line driving signal. The test set receiver is connected to a fault location pair; this is a voice frequency cable pair (loaded or nonloaded).

The pulse generator output consists of trios of pulses with a large quantity of bipolar violations. These are transmitted in specific patterns of positive trios (positive-negative-positive) and negative trios (negative-positive-negative). These positive and negative trios alternate at a voice frequency rate.



If the repeaters are operating properly, the trios will be regenerated and transmitted to the next repeater. A portion of the repeater output is coupled into the interrogation winding. This is filtered and connected to the fault location cable pair. At the central office, the fault location pair is measured for receive level at one of the twelve voice frequencies corresponding to filters F1 through F12. A faulty repeater is detected by no received signal or by a low receive level.

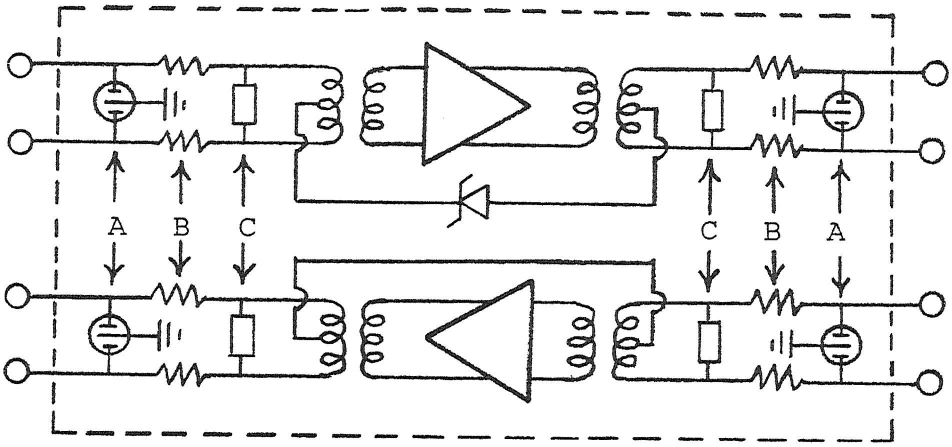
With this system, office repeaters (at both ends) and eleven line repeaters can be interrogated using one fault pair. Additional repeaters can be interrogated using additional fault pairs.

Other interrogation schemes utilize the same twelve frequencies and amplifiers in arrangements that provide for interrogation in both directions from one or both central offices (loop around interrogation); or arranged to interrogate up to 24 repeaters over one interrogation pair (in one direction).

When carrier system acceptance tests are made, it is important to include the interrogation system. The acceptance test data can be used as a benchmark to better interpret the fault location data at a later time.

Repeater Protection

Electrical protection of repeaters is generally provided in the following manner.



High voltage gap devices (A) are used to limit the voltage across the line terminals. These are usually 350 volt 2 element or 3 element gas tubes. Series resistors (about 5.6 ohms each) provide current limiting for lightning and electric system fault currents flowing through the repeater. Low voltage limiting devices (C) are placed across the input and output of the sensitive electronic circuits. PCM repeater protection has been much improved in recent years.

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